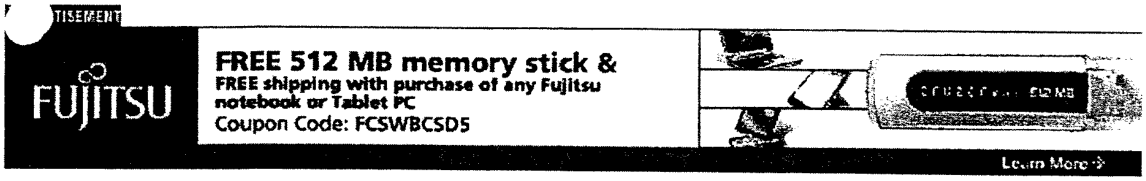


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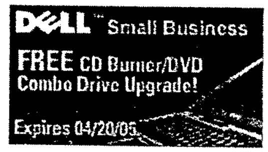
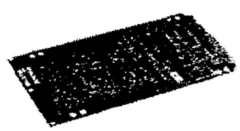
## Read product information for the Lucent (TNT-SL-48MODV3-S-C) Modem





Each Series56 Digital Modem module supports both analog and cellular connections at speeds up to 56 Kbps. It has 48-port Digital Modems (DM48) and occupies two expansion slots on the MAX TNT. Remote users with a modem and an analog or cellular line can dial into the MAX TNT over T1, DS3 or T1/E1/PRI lines.

See product details
> Write a review at Epinions.com

### Key Features

| | |
|---|---|
| Type | Modem (Digital) |
| Max. Transfer Rate | 56 Kbps |
| Connectivity Technology | Wired |
| Platform | PC |

### Protocols

| | |
|---|---|
| \nalog Modulation | ITU V.34, ITU V.32bis, ITU V.32, ITU V.17, ITU V.22bis, ITU V.29, ITU V.22, ITU V.90, ITU V.27ter, ITU V.21, ITU V.23, Bell 212A, Bell 103, K56Flex, ITU V.34bis, ITU V.33, Bell 212 |

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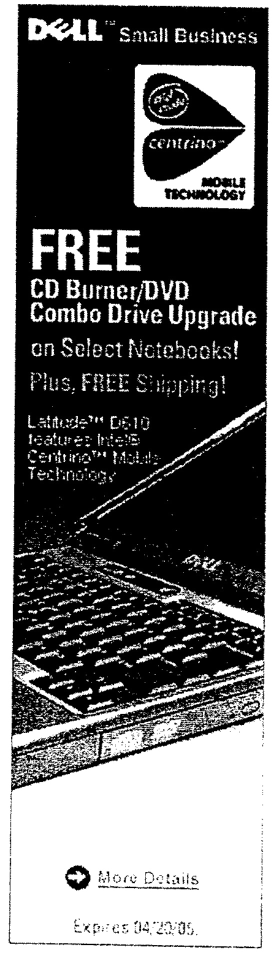
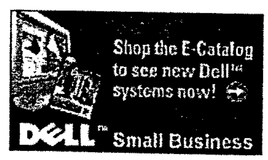
| | |
|---|---|
| Error Correction | ITU V.42, MNP-4, MNP-3, MNP-2, MNP-10, MNP-10EC |
| Data Compression | ITU V.42bis, MNP-5 |
| Digital Signaling | ISDN PRI |
| **Other Features** | |
| Max. Fax Transfer Rate | 14.4 Kbps |
| 56K Technology | V.90, K56Flex |
| **Dimensions** | |
| Depth | 1.62 in. |
| Height | 8.67 in. |
| Width | 10.99 in. |
| **Miscellaneous** | |
| Package Qty. | 1 |
| MPN | TNT-SL-48MODV3-S-C |
| Product ID | 20163035 |




### Additional resources







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## Read product information for the Lucent (TNT-SL-HDLC2) Expansion Module

This HDLC slot card has identical functionality to TNT-SL-HDLC2 with hardware support for encryption included (requires software option for encryption).

- See product details
- > Write a review at Epinions.com



| Key Features | |
|---|---|
| Form Factor | Plug-In Module |
| Interface | Expansion Slot |
| Connectivity | Cable |
| **Other Features** | |
| Package Qty. | 1 |
| **Miscellaneous** | |
| MPN | tnt-sl-hdlc2 |
| UPC | 000001224689 |
| Product ID | 22208116 |

### Additional resources



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# EXHIBIT 19

PC PCMAG.COM THE INDEPENDENT GUIDE TO TECHNOLOGY

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What are you shopping for? TNT SL CT1 in Expansion Modules Find

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## Read product information for the Lucent (TNT-SL-CT1) Expansion Module

This product is currently not available for purchase at any of the stores we search



Lucent gives its customers a unique opportunity to configure their network equipment the way the want it to be. By providing various network modules, Lucent satifies the needs of most demanding users and gurantees that its modules are fully compatible with their parent systems and deliver outstanding results.

See product details

> Write a review at Epinions.com

### Key Features

| | |
|---|---|
| Data Transfer Rate | 1.5 Mbps |
| Form Factor | Plug-In Module |
| Connectivity | Cable |
| Cabling Type | Network |
| latform | PC |

### Other Features

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d3

| | |
|---|---|
| Package Qty. | 1 |
| **Miscellaneous** | |
| MPN | TNT-SL-CT1 |
| Product ID | 20619944 |

**Additional resources**

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For Select Dell Latitude Dell
www.dell4me.com

DSX-1 Expansion Module
For the TSU Technology at Staples
www.Staples.com

Find It For Sale
Low Priced Expansion Modules. Huge Selection! (aff)
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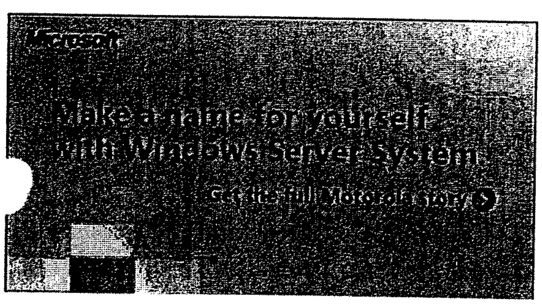
Kit [Expansion Module]
Matrix/Stacking Module Network- Modules at DirectDial
www.DirectDial.com

ebay Search for TNT SL CT1

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# EXHIBIT 20

# MCI Internet Dial Wholesale

MCI has developed one of the world's largest infrastructures for the carriage and provision of Internet services. is the capacity of our network, that we can also carry traffic for other Internet Service Providers (ISPs) or Online Service Providers (OSPs). You may be surprised to learn that a significant amount of MCI's revenue comes fro source, with Microsoft MSN On-line (AOL) being one of the most well known OSPs for whom we provide Whole Services. MCI has taken the provision of Internet services to new levels with its Wholesale Services: Internet D ISP Connect, Internet Dial VIP Managed Modem, Internet Dial VIP Access, and Internet Dial VIP Branded Dial. MCI's Wholesale Service has been rigorously improved and refined since its launch making it a tried and truste product - the ideal wholesale solution.

Today, more and more organisations are discovering the benefits of providing their customers with fixed and m dial-up Internet access. For traditional providers, such as Internet Service Providers (ISPs), Application Service Providers (ASPs), Online Service Providers (OSPs), and Mobile Operators, access is a primary source of rever For other organisations, including banks, retail chains and media owners, offering 'own brand' Internet connecti a value-added service that can boost customer loyalty and increase exposure.

With MCI Internet Dial Wholesale, you can join major players like CompuServe/AOL, EarthLink Networks, GTE The Microsoft Network (MSN), and offer high-quality, fixed and mobile dial-up Internet access via MCI's own gl Dial Access Network and Internet backbone. This is one of the easiest, most reliable and cost-effective ways to deliver dial-up Internet access to your customers.

By outsourcing network provision you are free to concentrate on meeting your customers' needs. Your busines processes are faster and more streamlined without the cost and resources required to build and maintain the network yourselves. You can take advantage of a leading-edge IP-based platform, with minimal investment anc effort, and offer new and exciting services to your clients. The MCI Internet Dial Wholesale service gives you ar your customers access to this network. A range of options is available so you can select which elements of the service you wish to outsource. Whatever edition you choose, you retain complete control of your branding and pricing.

## Advantages at a glance

**Quality**: Give your customers high-quality global Internet access and fast connect times using MCI's high performance network.

**Cost**: You gain low cost entry to this lucrative market. Your users benefit from lower call charges when oversea

**Revenue**: Create new sources of revenue and faster service roll-out using MCI products and services.

**Value-added**: Increase awareness of your company and brand.

**Control**: Flexible options mean you can select exactly which elements to outsource and design your marketing strategy to suit your business.

**Performance**: Shorter time to content provides a more compelling experience for users

**Low risk**: There is no need for investment in network infrastructure, nor network monitoring and network supen

## Features and benefits

As an MCI Internet Dial Wholesale customer, you are the Internet Service Provider. You direct the sales and marketing, brand the service as your own, and set the retail pricing. You are free to focus on the business of providing high-quality services to your customers, while MCI and our partners take care of the rest.

MCI's Network Operations Centres (NOCs), based in the USA, Europe and Asia-Pacific, closely monitor our glc network. Support is then provided around the clock to keep the network running at peak performance. By

maintaining our worldwide network to the highest standards we enable you to meet your customers' expectatio completely for Internet access.

MCI can provide a complete and integrated Internet service. Everything is covered from news through to email, (Domain Name Service), client software and personal Web space. We can even help you to select a specialist partner for end-user customer support*. If you prefer you can manage any of these elements yourself. The choi yours.

* End user support can be co-ordinated by MCI. We will introduce you to our high-calibre partners who can provide support major European languages.

We ensure proactive account management as part of the MCI Internet Dial Wholesale service. You will be assi dedicated technical support to work in partnership with your own operations department and ensure you servic deployment is as smooth as possible.

## Variants

MCI Internet Dial Wholesale is available in four different versions,

- Internet Dial ISP Connect
- Internet Dial VIP Managed Modem
- Internet Dial VIP Access
- Internet Dial VIP Branded Dial

## ISP Connect

The ISP connect service is the most basic ISP service and makes use of the voice interconnect network that M has in the UK. The calls, made by your end-users are terminated on your modem infrastructure. You will typical provide Modem indrastructure, Internet Backbone, email, DNS, news, Web and RADIUS servers, as well as en user support. As the wholesaler you will be able to call upon MCI's expert support resources.

## VIP Managed Modem

The VIP Managed Modem service gives you and your end-users access to MCI's world-class dial-up modem infrastructure in the UK. The traffic is tunneled through L2TP to your LNS This service is ideal for organisations a strong in house technical team that prefer not to invest in building and managing their own modem infrastruct and have internet backbone capacity (Global transit from MCI or another carrier). The service can be customise meet your requirements but you will typically provide Internet Backbone, email, DNS, news, Web and RADIUS servers, as well as end-user support. As the wholesaler you will be able to call upon MCI's expert support resou

## VIP Access

The VIP Access service gives you and your end-users access to MCI's world-class dial-up infrastructure in ove countries worldwide. This service is ideal for organisations with a strong in house technical team that prefer not invest in building and managing their own infrastructure. The service can be customised to meet your requirem but you will typically provide email, DNS, news, Web and RADIUS servers, as well as end-user support. As the wholesaler you will be able to call upon MCI's expert support resources.

## VIP Branded Dial

Designed for large organisations that want to provide a branded dial-up Internet service to a large customer ba

This brandable product is modular, comprising a number of different service options which can be chosen depe
on your individual requirements. **[more]**

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# EXHIBIT 21

Business Communications Review

September 1, 2002

**SECTION:** No. 9, Vol. 32; Pg. 54; ISSN: 0162-3885

**IAC-ACC-NO:** 91916299

**LENGTH:** 3104 words

**HEADLINE:** Traffic engineering for voice over IP.

**BYLINE: Hills, Michael** T.

**BODY:**

Although voice over IP (VOIP) is expected to replace conventional, circuit-switched voice traffic over the next few years, little work has been done so far to modify traditional voice traffic engineering for VOIP. This doesn't pose too much of a problem for carrier-class VOIP applications, in which conventionally originated and terminated PSTN voice calls are transported over IP "clouds" in the long-haul backbones. The real challenge comes in trying to engineer networks that will carry a mix of VOIP and other IP traffic flows.

The traffic-engineering implications of VOIP start with the fact that the Internet Protocol (IP) is not well suited to voice transmission. We have learned to accept the delays inherent in IP networks when, for example, we wait to watch a Web page being drawn. But two-way VOIP quality is not acceptable unless the delay is less than 150 milliseconds (ms) and the voice packets are played out at a uniform rate (i.e., jitter-free).

While we have many choices of packet voice and compression protocols, plus a variety of techniques to mark the voice packet headers for compression and other proprietary, special-handling options, the packet-voice equivalent of the circuit-switched voice Poisson and Erlang formulae have yet to be discovered (see p 56). In mixed IP traffic environments, VOIP engineers need to work with what information they have, use QOS controls and allow plenty of extra capacity.

The Basics: VOIP Coding And Compression

Transforming a standard, pulse-code-modulated (PCM) voice line—typically a 64-kbps DS0—into a stream of VOIP packets takes a number of steps. In PCM, an 8-bit voice sample is sent 8,000 times per second, or once every 0.125 ins. We could put each of these 8-bit samples in its own packet, but the additional header information that must accompany each packet would make this very inefficient.

There are three levels of packet header information needed to tell the IP network where to deliver the packet, where it came from, what kind of packet it is, and so forth:

* IP header with 20 bytes (160 bits).

* User Datagram Protocol (UDP) header with 8 bytes (32 bits).

* Real Time Protocol (RTP) header with 12 bytes (96 bits).

These headers are sent sequentially, ahead of the digitized voice sample, which is actually the payload of the RTP header. The result is 40-bytes (320 bits) of overhead for every packet, as shown in Figure 1. If we sent each PCM voice sample in its own packet at 8,000 packets per second, we would need 2.6 Mbps of bandwidth capacity to packetize a single DS0.

ATM adds yet another packet level, with an additional five-byte (40-bit) header for every 48 bytes, or 384 bits of

payload. The ATM payload includes the IP, UDP and RTP headers, plus the voice sample.

The first decision is obvious—put more voice samples in each packet. The bigger the packet, the fewer we have to send. But that also increases the delay between packets. Keeping this trade-off in mind, VOIP protocol designers have found that an interval of 30 ms (or 33.3 times per second) works well. Each sample still occurs every 0.125 ms, so in 30 ms we will have 240 samples. At 8 bits per sample this is 1,920 bits. With the 320-bit overhead, each packet will now be 2,240 bits, and the effective bandwidth requirement is only 75 kbps.

Table 1 shows the relevant characteristics of the most common voice-coding algorithms, as standardized by the ITU (International Telecommunication Union). G.711 is the ITU's PCM standard, while the G.729A algorithm was popularized by frame relay voice, and G.723.1 by Microsoft's Net Meeting application and the H.323 multimedia standards. Both G.729A and G.723.1 are considered equivalent in audio quality to a PSTN long-distance call, or "toll quality," especially when optional silence suppression, voice activity detection (VAD), comfort noise and other proprietary enhancements are added. Other standardized algorithms provide lower quality but can still be adequate depending on the application.

Other ways to reduce voice-packet overhead include compressing the headers and packing samples from multiple voice conversations into a single packet. There are various algorithms for doing this, and they can reduce overhead by as much as 30 percent. Since they are proprietary, however, both ends of the connection must agree on how they are used, which usually means matching equipment at each end.

## Engineering Backbone VOIP Links

In carrier-class VOIP, calls are collected and delivered as 64-kbps DS0 streams in DS1 or higher channelized systems. Converting them to VOIP for the long-haul transport saves local and long distance service providers a lot of money.

For example, suppose that we have 10,000 busy hour calls with an average holding time of 5 minutes per call that need to be carried between New York and Los Angeles. Using classic traffic engineering, this means we have 833.3 Erlangs of traffic (or 30,000 CCS) and will need 862 DS0 trunks to carry it at a P01 grade of service level. Given that 862 trunks equates to 35 DS1s, two DS3s would probably be implemented.

Now assume we also have an additional 20,000 calls from New York to Chicago joining the coast-to-coast load. This would add another 1,696 DS0s or 71 DS1s (probably implemented as 3 DS3s) for the New York—Chicago leg of the route.

To convert the total New York—Chicago load to G.723.1-encoded packet voice, we take the number of DS0s needed (862 + 1,696 = 2,558) and multiply that by the amount of IP bandwidth needed to carry the load using G.723.1 (2,558 channels x 17.1 kbps per channel = 43,741,800 bps, or 43.7 Mbps). We then divide by the speech factor, which is about 50 percent (both parties on the call are assumed to take turns speaking). Now we have a bandwidth load of 21.8 Mbps. Here is the formula: *IP Bandwidth Required = Peak channels required X IP bandwidth X Speech factor.

Rather than the five DS3s needed to carry the traditional voice load between New York and Chicago, we now need only one. This is the first, basic attraction of VOIP for carriers, although the second is almost as powerful: Rather than linking nodes in expensive meshes and tandems of dedicated circuits, carriers can use routed IP networks that let every node simply throw its VOIP traffic into the cloud. Of course, they have to make sure the cloud performs adequately, so that the VOIP packets come out where they are supposed to, and within the ITU's 150-ms delay requirement for satisfactory speech.

Once the calculations above are made, it's easy to see why most carriers have adopted this flavor of VOIP—they can save anywhere from half to two-thirds of their backbone bandwidth, and perhaps reduce their spending on carrier equipment. The example assumes, however, that only VOIP traffic is riding on the IP network. What happens to VOIP when other IP traffic flows, such as email and Web surfing and file transfers, have to share the bandwidth capacity?

## Mixed Traffic Is Tough

To visualize a mixed IP traffic network, imagine a VOIP traffic engineer at the top of Niagara Falls, where he must place a series of ping-pong balls (the VOIP packets) into the water (the flow of regular IP traffic) and guarantee that they all arrive at the bottom, in sequence and with a small, fixed time delay between each ping-pong ball. Needless to say, this is extremely difficult.

In the early 1990s, Bell Lab researchers discovered that Internet traffic couldn't be characterized by classical means.

It does not arrive according to Poisson's distribution formula, and it does not endure according to Erlang C's steady-state assumption or exponential holding-time formula. This is hardly surprising, when you consider this type of traffic is made up of millions of little transactions as emails and Web pages are sent and retrieved, plus long transactions as people download large files, or audio and video streams.

Although the Bell Lab researchers started out thinking they could model Internet traffic with Erlang-C, perhaps buffering traffic at each router to smooth the load, it turned out that nothing was further from the truth. What the researchers found is that IP network traffic exhibits a bursty, fractal pattern. They called this "self-similar traffic," because the distribution looks exactly the same over any given time scale—two weeks, two days, two tenths of a minute, etc. In practical terms, however, the self-similar quality of IP traffic makes the traditional traffic engineering equations irrelevant.

Self-similar traffic has two primary traits:

* The duration of a transmission depends on its prior length: In other words, a connection that has been in existence for a long time will most likely continue to exist.

* Things don't average out: The reason busy-hour calculations are effective in traditional voice networks is because the traffic averages out over a sufficiently long period. But Internet traffic doesn't behave like this, it remains unpredictably bursty no matter how long you watch it.

Now What Do We Do?

In networks devoted to VOIP traffic, we can expect Poisson and Erlang to apply, and we can use the capacity calculations above. But, so far, we do not have a good way to characterize mixed IP traffic; That means we can't perform the kinds of capacity versus service-level calculations on mixed IP traffic that Poisson and Erlang allow us to do in traditional voice networks.

Most of the efforts that are called VOIP traffic engineering today are really nothing more than attempts to ensure that the voice packets are delivered in a timely manner, whatever the state of the IP network. This can be done in two ways: By segregating the VOIP traffic from the other flows, or by marking VOIP packets for priority handling via class and quality of service (COS and QOS) mechanisms, although each router along the path must use the same mechanisms for the techniques to be effective.

Quite a lot of material is available to help guide VOIP engineers as they plan to add packet voice to their IP data LANs and WANs (for example, see www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/solution/3_pla nni.htm#xtocid48). The basic advice given is to:

* Measure the peak occupancy of the links in the existing data network.

* Use 5-minute averages of the measured peaks. Size the mixed network based on this, plus the calculated voice traffic load. If the new peaks in usage exceed 50 percent of the capacity, add more bandwidth.

* Segregate VOIP traffic onto its own portion of the network (virtual LAN, subnet) and use COS and QOS protocols to guarantee preferential treatment for VOIP.

* Use traffic shaping to make sure no source can swamp the network and prevent VOIP (or other) packets from getting through.

There is considerable research activity in progress to develop methods of characterizing IP traffic and to develop the new traffic engineering tools needed to perform more precise designs (see references at www.htlt.com/articles). As yet, however, nothing with the ease of use of the Erlang tables has emerged.

Conclusion

What does this mean for enterprise telecom/network managers? First, it seems that VOIP is definitely in everyone's future, if not already on the planning table. IP PBXs, IP call centers and, perhaps even multimedia services will eventually displace their circuit-switched predecessors.

It remains to be seen, however, whether most of these installations will be mixed P traffic environments, or if VOIP will be kept separate from other P traffic. In either case, as you plan for these systems, you will need to learn about the IP network mechanisms which can affect the quality of VOIP calls. This will probably involve exchanging information with others in the IT department, many of whom may need you to help them understand the special requirements of voice

traffic. Tools that you can use to measure P traffic in ways that will be useful for network capacity planning are only just appearing and, as discussed above, we are awaiting the modeling formulae that will make these exercises consistent and predictable.

TABLE 1

Characteristics Of Common Voice Coding Algorithms

| Coding Algorithm | | Bandwidth | Voice Sample Frequency | IP Bandwidth |
|---|---|---|---|---|
| G.711 | Standard PCM | 64.0 kbps | 0.125 ms | 2,624.0 kbps |
| | | | 30 ms | 75.0 kbps |
| G.723.1 | MP-MLQ | 5.3 kbps | | 16.0 kbps |
| | ACELP | 6.3 kbps | 30 ms | 17.1 kbps |
| G.729A | CS-ACELP | 8.0 kbps | 10 ms | 20.0kbps |

FIGURE 1

A Voice Over IP Packet

| | Byte 1 | Byte 2 | Byte 3 | Byte 4 |
|---|---|---|---|---|
| 1-4 | IP | | | |
| 5-8 | IP | | | |
| 9-12 | IP | | | |
| 13-16 | IP | | | |
| 17-20 | IP | | | |
| 21-24 | UDP | | | |
| 25-28 | UDP | | | |
| 29-32 | RTP | | | |
| 33-36 | RTP | | | |
| 37-40 | RTP | | | |
| 41- | digitized vocie | | | |

FIGURE A

Voice Traffic Engineering Time Line

1880 Telephone invented.

1890 Poisson distribution applied horse
kick deaths

1900 Erlang discovers telephone calls
follow Poisson.
Erlang-B formula discovered.

1910 Erland-C formula discovered.

1920

1930

1940 Mike Hills born

1950 TAT-1 transatlantic cable

1960 Kleinrock, Baron and Davies publish
important packet switching
theories.

1970 ITU standardizes PCM.

1980 TCP/IP established. WWW introduced.

1990 Self-similar nature of
internet traffic discovered.
U.S. telephone companies ask
Congress to ban VOIP.

2000 H.323 adapted for VOIP.
VOIP traffic expected to
exceed TDM voice.

2010 Next Erlang will emerge.

RELATED ARTICLE: Poisson And Erlang

Although Internet traffic—email, Web surfing, file transfers and downloads—does not adhere to the classical voice traffic engineering models, current voice traffic still arrives in keeping with Poisson distribution and endures according to Erlang-C. In fact, the high quality of classic voice telephony owes quite a lot to the work of French mathematician Simeon Poisson (1781-1840), and Danish mathematician Agner Kraruo Erlang (1878-1929). Figure A shows key events in the history of voice traffic engineering.

Poisson studied ways to characterize random events, and he developed what we call the Poisson distribution, which simply describes the average number of events that you expect in a given period. Perhaps the most famous application of this theory was its use to model the deaths by horse kicks in the Prussian army between 1875 and 1894. Russian mathematician Ladislau Bortkiewicz (1868-1931) calculated that the 200 deaths in the 10 corps over a twenty-year period worked out to between 0 and 4 deaths per corps per year. This was thought to be statistically random enough not to require a more sinister explanation.

In 1908, when the Danish mathematician Erlang went to work for the Copenhagen Telephone Company, he set out to solve the key problem in telephone network design: How many trunks are needed to carry a given amount of calling? His major discovery (published in 1909) was the proof that telephone calls distributed at random follow Poisson's law of distribution. By estimating the average number of calls expected within an hour, Erlang found he could use Poisson's law to predict the probability that the actual number arriving in a future hour would exceed the number of trunks available. He developed the Poisson traffic tables, which show the relationship between the number of trunks and the probability of blockage (more calls than trunks).

Erlang refined his model by assuming that blocked calls disappear and never reappear, which is reflected in his Erlang-B equation. In 1917 he extended this formula to account for situations in which the caller would wait in an orderly queue if a trunk or operator was busy at the instant that they initiated the call. This formula, called Erlang-C, required two new assumptions:

* Telephone call durations are exponentially distributed. In simple terms, this says the probability that a call will terminate is independent of how long the call has already been in progress. This unlikely assumption was found to be true in the early 1900's and is still true today, for voice calls.

* Steady-state assumption: This says that the calling rate (the speed with which new calls arrive in the networks) has been in effect for a long time before the period being examined. This turns out to be valid for classic voice telephony traffic, in which calls average 5 minutes, relatively short compared to the length of the usual observation period (30 or 60 minutes).

Long distance and local voice traffic can still be modeled using Erlang-B. But neither Internet traffic, nor people who are using dialup Internet access obey the rules of Poisson or Erlang. Internet traffic itself is bursty and unpredictable, unlike the smooth and repetitive voice calling loads, while people tend to stay on Internet access "calls" much longer than the average 5-minute voice calls. So we cannot use the classic equations either for Internet traffic or for Internet access.

Dr. **Michael** T. **Hills** is founder and president of HTLT, leaders in telecom cost containment systems. Dr. Hills has developed a range of mathematical approaches to design networks economically and efficiently Dr Hills can be reached at 301/362-9404 or mhills@htlt.com.

**IAC-CREATE-DATE:** September 30, 2002

**LOAD-DATE:** October 01, 2002

# EXHIBIT 22

# Request for Payment of Internal Revenue Taxes

**(Bankruptcy Code Cases—Administrative Expenses)**

Department of the Treasury/Internal Revenue Service

United States Bankruptcy Court for the SOUTHERN
District of NEW YORK STATE

**In the Matter of:** UUNET TECHNOLOGIES, INC.
500 CLINTON CENTER DRIVE
CLINTON, MS 39056

Fiduciary:

Amendment No. 1 to Request for Payment dated 02/25/2004

Case Number
02-42302-AJG

Type of Bankruptcy Case
CHAPTER 11

Date of Petition
07/21/2002

Filed: USBC - Southern District of New York
Worldcom, Inc., Et Al.
02-13533 (AJG)
0000038365

1. The undersigned, whose business address is IRS Insolvency Group 4 290 Broadway Stop 5TH FLR New York, NY 1000,/is the agent of the Department of the Treasury, Internal Revenue Service, and is authorized to make this request for payment on behalf of the United States.
2. Request is made for payment of taxes and any interest or penalty due under the internal revenue laws of the United States, as shown below.
3. The ground of liability is taxes due under the internal revenue laws of the United States.

**Administrative Claims**

| *Taxpayer ID Number* | *Kind of Tax* | *Tax Period* | *Tax Due* | *Interest Due* | *Penalty Due* | *Balance Due* |
|---|---|---|---|---|---|---|
| 54-1543611 | 5 EXCISE | 12/31/2002 | $3,997,592.00 | $275,259.66 | $0.00 | $4,272,851.66 |
| 54-1543611 | 5 EXCISE | 03/31/2003 | $3,737,766.00 | $208,960.20 | $0.00 | $3,946,726.20 |
| 54-1543611 | 5 EXCISE | 06/30/2003 | $1,726,848.00 | $73,705.70 | $0.00 | $1,800,553.70 |
| 54-1543611 | 5 EXCISE | 09/30/2003 | $1,395,081.00 | $42,549.48 | $0.00 | $1,437,630.48 |
| 54-1543611 | 5 EXCISE | 12/31/2003 | $1,476,730.00 | $29,788.86 | $0.00 | $1,506,518.86 |
| 54-1543611 | 5 EXCISE | 03/31/2004 | $2,466,803.00 | $23,088.91 | $0.00 | $2,489,891.91 |
| 54-1543611 | 5 EXCISE | 04/01/2004-04/20/2004 | $822,268.00 | $0.00 | $0.00 | $822,268.00 |
| | | | $15,623,088.00 | $653,352.81 | $0.00 | $16,276,440.81 |

**Total Amount Due: $16,276,440.81**

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U.S. BANKRUPTCY COURT
S. DIST. OF NEW YORK

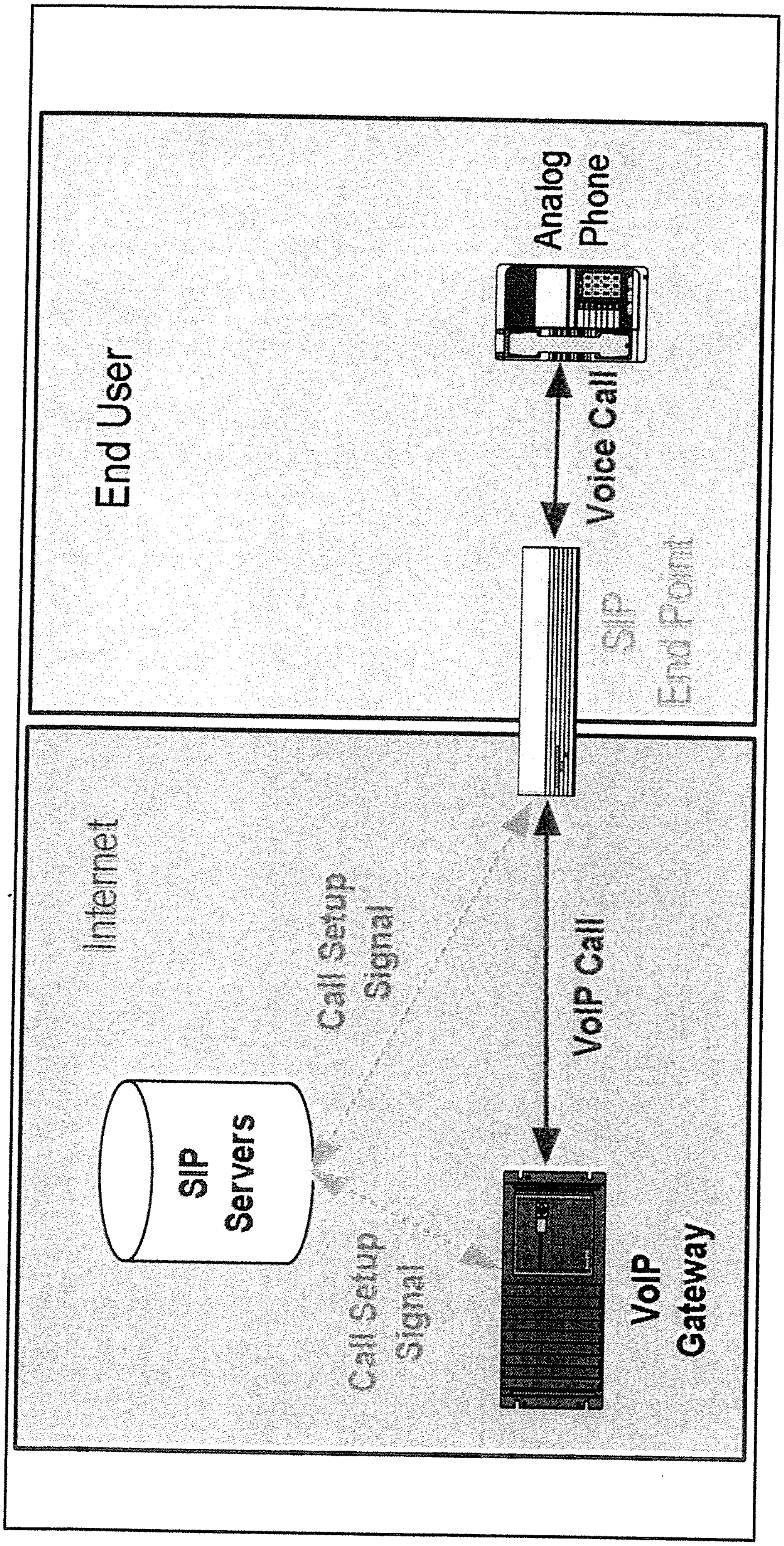
*5 UNASSESSED TAX CLAIMS HAVE BEEN FILED DUE TO PROPOSED ADDITIONAL ASSESSMENT DUE TO AN EXAMINATION OF THE TAX RETURN FOR THE PERIOD UNASSESSED.*

The amount due includes interest and penalty computed to 07/09/2004. Compound interest will accrue at the rate established under IRC Section 6621(a) and late payment penalty will be charged under IRC Section 6651. If the claim is paid after 07/09/2004, contact NAN DILLINGHAM at (212) 436-1341 for the current balance.

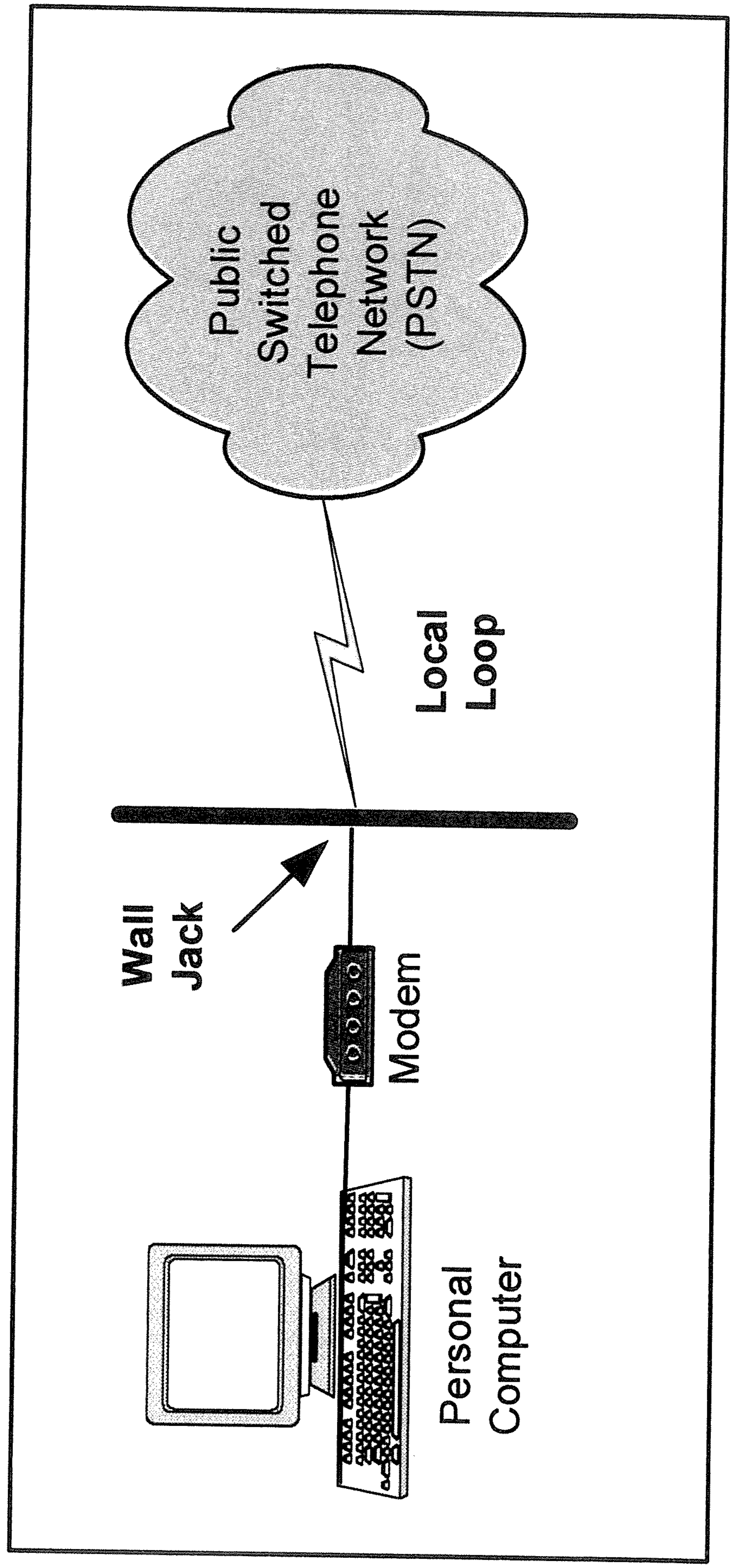
| Penalty for Presenting Fraudulent Claim - Fine of not more than $5,000 or imprisonment for not more than 5 years or both - Title 18, U.S.C., Section 152. | Signature N. Dillingham / Marcia Smith | Date 06/28/2004 |
|---|---|---|
| | Title Insolvency Territory 2 Manager | Telephone Number (212) 436-1341 |

Form 6338 - A (C)

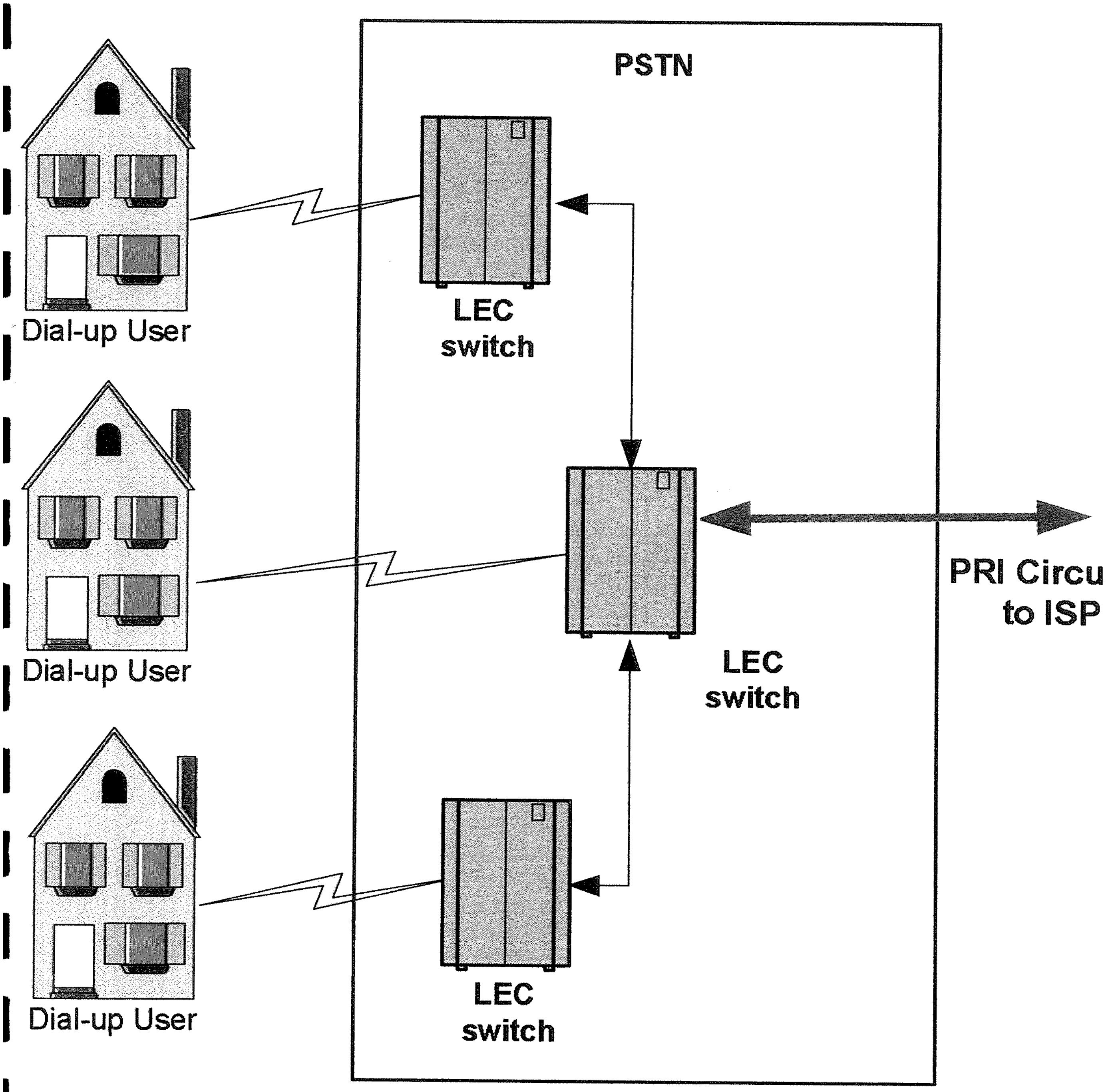
# EXHIBIT 23


End User
Analog Phone
Voice Call
SIP End Point
Internet
Call Setup Signal
VoIP Call
SIP Servers
Call Setup Signal
VoIP Gateway

# EXHIBIT 24

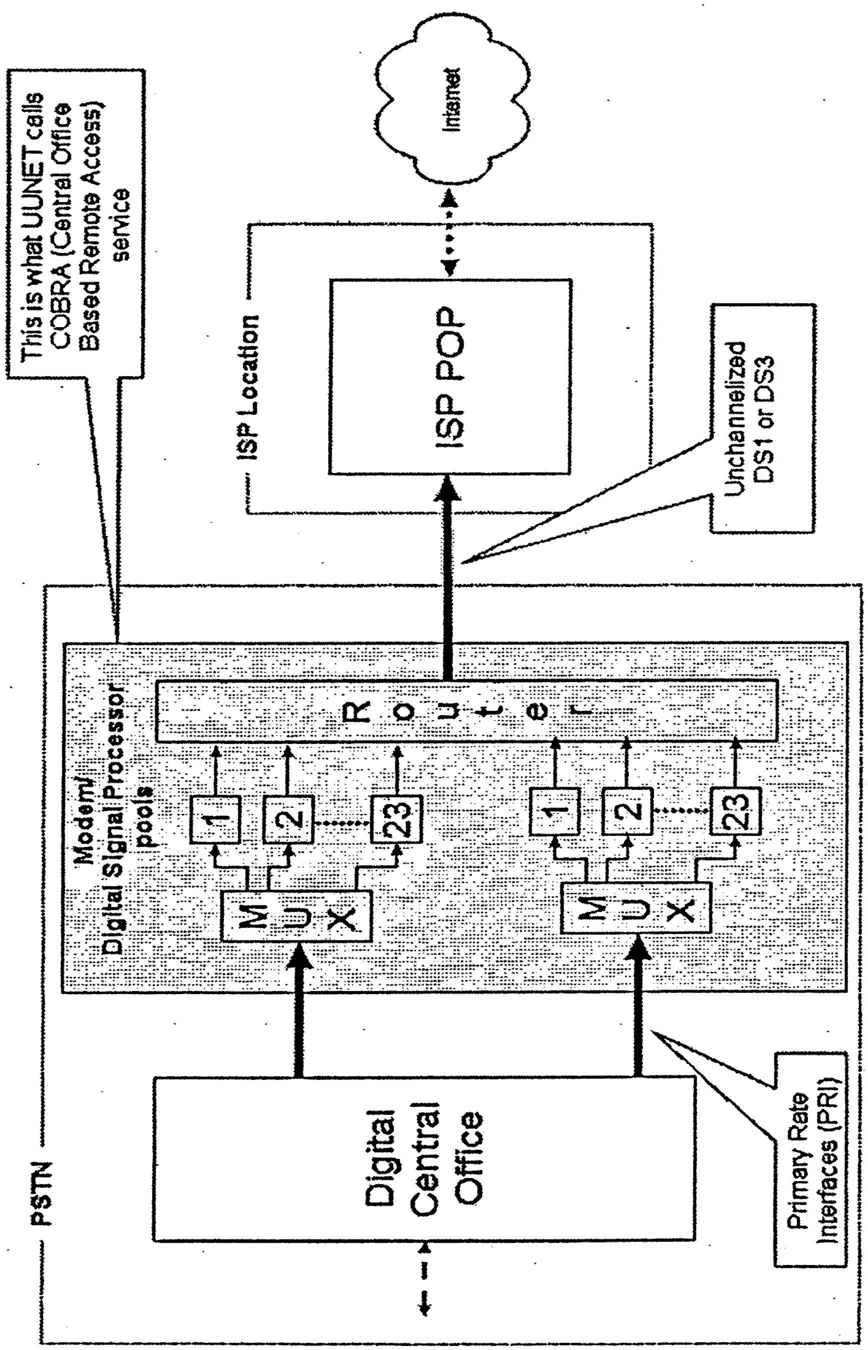

Public
Switched
Telephone
Network
(PSTN)
Local
Loop
Wall
Jack
Modem
Personal
Computer

# EXHIBIT 25


PSTN
LEC switch
Dial-up User
LEC switch
PRI Circuits to ISP
Dial-up User
LEC switch
Dial-up User

# EXHIBIT 26

Colocate Modem/Routers in Central Office




Typically 6 PRIs can be concentrated onto 1 DS1

Figure 5

# CONFIDENTIAL –FOR *IN CAMERA* REVIEW ONLY PURSUANT TO BANKRUPTCY COURT PROTECTIVE ORDER

**UNITED STATES BANKRUPTCY COURT**
**SOUTHERN DISTRICT OF NEW YORK**

| | | |
|---|---|---|
| **In re** | : | |
| | : | **Chapter 11** |
| **WORLDCOM, INC., et al.,** | : | |
| | : | **Case No. 02-13533 (AJG)** |
| | : | |
| **Debtors.** | : | **(Jointly Administered)** |

**THESE EXHIBITS ARE SUBJECT TO A PROTECTIVE ORDER OF THE BANKRUPTCY COURT, DATED OCTOBER 5, 2005 (DOCKET NO. 17290). ACCORDINGLY, THESE EXHIBITS ARE FOR *IN CAMERA* REVIEW ONLY, SHALL BE TREATED AS CONFIDENTIAL AND SHALL NOT BE SHOWN TO ANY PERSON OTHER THAN THOSE PERSONS DESIGNATED IN THE PROTECTIVE ORDER.**

| EX. NO. | DESCRIPTION | MARKED | OFFERED | OBJECT | ADMIT |
|---|---|---|---|---|---|
| 27. | Verizon CyberPOP Agreement<br><br>Verizon letter agreement (DOJ00002 – DOJ00018)<br>Amended and Restated RAS Equipment Resale Agreement (DOJ00212 – DOJ00217)<br>Amended and Restated Exhibit B (DOJ00235)<br>Amended and Restated Exhibit A (DOJ00245)<br>Letter of GTE (DOJ00133 – DOJ00140) | | | | |
| 28. | SBC COBRA Agreement<br><br>Master Services Agreement Between SBC Global Services, Inc. and UUNET Technologies, Inc. (DOJ00119 - DOJ0132)<br>Cobra Architecture White Paper – Lucent/Ascend Equipment (DOJ00173 – DOJ00190) | | | | |

| EX. NO. | DESCRIPTION | MARKED | OFFERED | OBJECT | ADMIT |
|---|---|---|---|---|---|
| 29. | QWEST COBRA Agreement<br><br>Amended and Restated Master Services Agreement (DOJ00019 - DOJ00028)<br>Uniform Access Services Schedule (DOJ00029 – DOJ0034)<br>Amended and Restated Central Office-Based Remote Access Service Schedule (DOJ00035 – DOJ00045)<br>UUNET/Qwest COBRA Service Schedule (DOJ00218 – DOJ00222) (DOJ00234, DOJ00246) | | | | |
| 30. | BELLSOUTH RAS Agreement<br><br>Master Contract Services Arrangement for RAS Service (DOJ00052 – DOJ00057)<br>Attachment No. 1 to Master Contract Service Arrangement No. GA00-5399-00 (DOJ00058 – DOJ00068)<br>Amendment No. 1 to Dial Access Platform Service Agreement (DOJ00069 – DOJ00080)<br>Bell South Telecommunications Terms & Conditions for Dial Access Platform Service (DOJ00081 – DOJ00094)<br>UUNET COBRA 3COM Racked Equipment specifications Rev. 2, 8/21/00 (DOJ00223 – DOJ00233) | | | | |
| 31. | U S West Network Service Agreement General Terms and Conditions for Signaling System 7 Gateway Service/Central Office-Based Remote Access Service (DOJ00097 – DOJ00118) | | | | |
| 32. | ICG COBRA Agreement<br><br>Amended & Restated Remote Access Services Agreement (DOJ00141 – DOJ00159)<br>Service Description Lucent/Ascent TNT Equipment (DOJ00191 – DOJ00210) | | | | |

# EXHIBIT 27

ORIGINAL

COPY

verizon

Virginia P. Ruesterholz
Senior Vice President
Wholesale Services

Verizon Communications
1095 Avenue of the Americas, Room 4039
New York, NY 10036

Phone 212.395.1069
Fax 212.597.2563
Virginia.P.Ruesterholz@verizon.com

May 15, 2001

VERIZON & WORLDCOM CONFIDENTIAL

Louis R. Prestwood
Senior Vice President
Network Financial Management
MCI WORLDCOM Network Services, Inc.
2270 Lakeside Boulevard
Richardson, TX 75081

Dear Louis,

Our teams have completed their work on restructuring Verizon's delivery of modem aggregation services to MCI WORLDCOM Network Services, Inc. and its affiliated companies ("WorldCom" or "WCOM"), and on other matters related to the broader scope of business between our two companies. Verizon, which for purposes of this letter and the related documents, means the Verizon operating telephone companies listed on Exhibit 1, is ready to file new tariffs for the modem aggregation service, which we have named CyberPOP™ service. This letter sets out our agreements concerning the proposed tariff, additional administrative and procedural terms related to provision of CyberPOP service, and other business arrangements.

WorldCom agrees to pay Verizon all amounts owed (and not otherwise in dispute) with respect to existing CyberPOP and other Internet dial-up services that are the subject of this letter prior to Verizon filing the new CyberPOP service tariffs. In addition, Verizon and WorldCom have settled certain disputes relating to various charges, which settlement is memorialized in a separate Settlement Agreement executed concurrently with this letter agreement.

Subject to receipt of the payment of amounts owed, Verizon will file the tariffs no later than the next business day following the receipt of payment. The tariffs will be filed in the applicable federal interstate access tariff for each of the Verizon companies, and will be in accordance with the tariff set out at Exhibit 2.

This letter is WorldCom's subscription to the CyberPOP service under the new tariff.

For consistency, defined terms used in this letter have the same meaning as in the tariff. For purposes of this arrangement, WorldCom affiliates shall be deemed to include all entities controlling, controlled by, or under common control with WorldCom.

The following additional provisions apply to Verizon's provision of, and WorldCom's use of, the CyberPOP service and other business arrangements.